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APPLICATION NO.	FILING DATE	FIRST NAMED INVENTOR	ATTORNEY DOCKET NO.	CONFIRMATION NO.
09/745,387	12/21/2000	David R. Oran	2705-126	1119

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EXAMINER	
SEFCHECK, GREGORY B	

ART UNIT	PAPER NUMBER
2616	

DATE MAILED: 08/23/2006

Please find below and/or attached an Office communication concerning this application or proceeding.

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Office Action Summary	Application No.	Applicant(s)	
	09/745,387	ORAN, DAVID R.	
	Examiner	**Art Unit**	
	Gregory B. Sefcheck	2616	

-- The MAILING DATE of this communication appears on the cover sheet with the correspondence address --

Period for Reply

A SHORTENED STATUTORY PERIOD FOR REPLY IS SET TO EXPIRE 3 MONTH(S) OR THIRTY (30) DAYS, WHICHEVER IS LONGER, FROM THE MAILING DATE OF THIS COMMUNICATION.

- Extensions of time may be available under the provisions of 37 CFR 1.136(a). In no event, however, may a reply be timely filed after SIX (6) MONTHS from the mailing date of this communication.
- If NO period for reply is specified above, the maximum statutory period will apply and will expire SIX (6) MONTHS from the mailing date of this communication.
- Failure to reply within the set or extended period for reply will, by statute, cause the application to become ABANDONED (35 U.S.C. § 133). Any reply received by the Office later than three months after the mailing date of this communication, even if timely filed, may reduce any earned patent term adjustment. See 37 CFR 1.704(b).

Status

1)☒ Responsive to communication(s) filed on *23 May 2006*.

2a)☒ This action is **FINAL**. 2b)☐ This action is non-final.

3)☐ Since this application is in condition for allowance except for formal matters, prosecution as to the merits is closed in accordance with the practice under *Ex parte Quayle*, 1935 C.D. 11, 453 O.G. 213.

Disposition of Claims

4)☒ Claim(s) *1-13 and 15-57* is/are pending in the application.

4a) Of the above claim(s) _____ is/are withdrawn from consideration.

5)☒ Claim(s) *13 and 15-47* is/are allowed.

6)☒ Claim(s) *1-12,48-53 and 55-57* is/are rejected.

7)☒ Claim(s) *54* is/are objected to.

8)☐ Claim(s) _____ are subject to restriction and/or election requirement.

Application Papers

9)☐ The specification is objected to by the Examiner.

10)☐ The drawing(s) filed on _____ is/are: a)☐ accepted or b)☐ objected to by the Examiner.

Applicant may not request that any objection to the drawing(s) be held in abeyance. See 37 CFR 1.85(a).

Replacement drawing sheet(s) including the correction is required if the drawing(s) is objected to. See 37 CFR 1.121(d).

11)☐ The oath or declaration is objected to by the Examiner. Note the attached Office Action or form PTO-152.

Priority under 35 U.S.C. § 119

12)☐ Acknowledgment is made of a claim for foreign priority under 35 U.S.C. § 119(a)-(d) or (f).

a)☐ All b)☐ Some * c)☐ None of:

1.☐ Certified copies of the priority documents have been received.

2.☐ Certified copies of the priority documents have been received in Application No. _____.

3.☐ Copies of the certified copies of the priority documents have been received in this National Stage application from the International Bureau (PCT Rule 17.2(a)).

* See the attached detailed Office action for a list of the certified copies not received.

Attachment(s)

1) ☒ Notice of References Cited (PTO-892)
2) ☐ Notice of Draftsperson's Patent Drawing Review (PTO-948)
3) ☐ Information Disclosure Statement(s) (PTO-1449 or PTO/SB/08) Paper No(s)/Mail Date _____.
4) ☐ Interview Summary (PTO-413) Paper No(s)/Mail Date. _____ .
5) ☐ Notice of Informal Patent Application (PTO-152)
6) ☐ Other: _____.

DETAILED ACTION

- Applicant's Amendment filed 5/23/2006 is acknowledged.
- Claims 1, 10-13, 15, 17, 20, 22, 24, 33, 34, 36, 37, and 48 have been amended.
- Claim 14 has been cancelled.
- Claims 51-57 have been added.
- Claims 1-13 and 15-57 are pending.

Claim Rejections - 35 USC § 103

1. The following is a quotation of 35 U.S.C. 103(a) which forms the basis for all obviousness rejections set forth in this Office action:

> (a) A patent may not be obtained though the invention is not identically disclosed or described as set forth in section 102 of this title, if the differences between the subject matter sought to be patented and the prior art are such that the subject matter as a whole would have been obvious at the time the invention was made to a person having ordinary skill in the art to which said subject matter pertains. Patentability shall not be negatived by the manner in which the invention was made.

2. Claims 1, 2, 4-12, 50, and 56 are rejected under 35 U.S.C. 103(a) as being unpatentable over Kung et al. (US006775267B1), hereafter Kung, in view of Havens (US006735175B1).

- In regards to Claims 1, 2, 5, 10, 11, 50, and 56,

Kung discloses a method and software for controlling a voice-over-IP (VOIP) call system (Figs. 1-4; Col. 7, lines 15-25; Col. 8, lines 9-13; claim 1 – method for controlling a VOIP call).

Kung shows that VOIP call packets traveling through the IP network may be given a priority to maintain certain QoS requirements (Col. 7, lines 21-25; claim 1 – tracking adaptation scheme settings used for transmitting packet in a VOIP call).

Kung discloses the ability to change quality of service, required bit rate, priority, cost, etc. in real time in response to user input (Col. 7, lines 27-30; claim 1 – monitoring a user response/input that requests a different level of user perceived sound quality for the VOIP call; claim 56 – selecting call parameter according to maximum cost selection).

Kung discloses that calls may be initially conducted at a user's default settings of quality, cost, etc. (Abstract; Col. 28, lines 12-19; claim 2,25 – initially transmitting packets of VOIP call using best effort).

Kung discloses that the system is flexible so that a given communication can be dynamically altered according to customer preferences such as a user's desired quality of service. As such, the adaptation scheme setting are tracked and varied by the user (telephone endpoint, source or destination of call), though the user's input is processed at the central station (Col. 7, lines 27-35). The call settings may then be dynamically altered based on user input, requiring a call manager to reserve the necessary resources (Col. 30, lines 25-30; claim 1 – updating scheme settings during the call; claim 2,25 – monitoring the user response for a request to increase sound quality; claim 2,25 – requesting reservation of resources during the call when the increased sound quality request is detected prior to the reserved resources being used during the call and without necessarily using the entire requested resources during the call).

Kung further discloses that the real time changes to the VOIP call may be flexibly performed with regard to congestion in the network (Col. 7, lines 30-35; Col. 17, lines 55-59; claim 10 – monitoring congestion in a network used for conducting the call and varying adaptation schemes according to the user response and the monitored congestion).

Though Kung shows that quality of service control through a multimedia gateway control protocol may include codec choice (Col. 14, lines 59-65), Kung does not explicitly disclose determining and updating the preferred encoding algorithm based on user request for different sound quality during the VOIP call, at a telephone endpoint that is the source of the VOIP call.

Havens discloses changing quality of service for voice over IP calls. Havens shows implementing requested changes to quality of service by adjusting performance of the codec module (Fig. 2; Abstract; Col. 2, lines 31-43; Col. 4, lines 23-30). Havens shows that the call-originating user dials a change of QoS on the telephone when the perceived audio quality of the call is insufficient. Havens shows that these changes may be done in real time during the call (Col. 4, lines 12-53; claim 1 –telephone endpoint is originating source of call; claim 1 - determining and updating the preferred encoding algorithm based on user request for different sound quality during the VOIP call; claim 5 – increasing voice coder performance or reducing payload size after the resources are reserved; claim 11 – varying codecs used for encoding audio signals into digital data making up the packets; claim 50 – listening to audible signal after

dynamically varying adaptation schemes to determine level of user perceived sound quality, further varying the schemes to improve the audible signal when the perceived sound quality is low, and further listening to the improved signal to determine a level of user perceived sound quality).

It would have been obvious to one of ordinary skill in the art at the time of the invention to modify the method of Kung by adjusting coder performance in response to user requested change in quality of service, as shown by Havens. This would enable quality of service to be dynamically adjusted during a call without requiring changes to the bandwidth of the call.

- In regards to Claims 4,

Kung discloses a method for controlling a voice-over-IP (VOIP) call system that covers all limitations of the parent claims.

Kung shows that user input for changing the call quality may be performed before as well as during the call (Col. 7, lines 27-35; Col. 30, lines 25-30; claim 4 – conducting the already established call using reserved resources when the reservation request is accepted and the user response requests additional increases in sound quality).

- In regards to Claims 6 and 8,

Kung discloses a method for controlling a voice-over-IP (VOIP) call system that covers all limitations of the parent claims.

Kung discloses changing call parameters is accomplished through user input on a user device, such as the screen portions shown in Figs. 7-9. Kung discloses that user input may be collected via touchscreen (graphical user interface; Col. 20, lines 51-55; claim 6 – using a signal generated by an input device to detect the user response during the call; claim 8 – using a graphical user interface as the input device).

- In regards to Claims 7 and 9,

Kung discloses a method for controlling a voice-over-IP (VOIP) call system that covers all limitations of the parent claims.

Telephone units are also shown to be connected to the system for use as an input device by the user, including DTMF sensing logic (Fig. 3; Col. 23, lines 45-51; claim 7 – including using a dial or buttons on a telephone as the input device; claim 9 – including decoding DTMF signals to detect the user response).

- In regards to Claims 12,

Kung discloses a method for controlling a voice-over-IP (VOIP) call system that covers all limitations of the parent claims.

Referring to Fig. 9B, Kung shows that user input for changing call parameters may include cost icons (claim 12 – detecting a user response selecting a cost for the VoIP call and varying the adaptation schemes according to the selected cost).

3. Claims 3 is rejected under 35 U.S.C. 103(a) as being unpatentable over Kung in view of Murphy et al. (US006282192B1), hereafter Murphy.

- In regards to Claim 3,

Kung discloses a method for controlling a voice-over-IP (VOIP) call system that covers all limitations of the parent claims.

Kung does not explicitly disclose utilizing an RSVP request during the call to request reservation of resources.

Murphy discloses a routing scheme in a packet switched network for processing voice over IP calls (Title; Abstract). Murphy shows that call admissions control protocols, such as RSVP, can be used for establishing VOIP calls (Col. 8, lines 30-43; Col. 9, lines 54-56; claim 3 – requesting reservation of resources comprises making RSVP request during the call).

It would have been obvious to one of ordinary skill in the art at the time of the invention to utilize RSVP requests for reserving resources, as shown by Murphy, during a VOIP call in the method of Kung. This modification would enable a bandwidth reservation request for the call to specify certain quality of service requirements needed to improve the sound quality solicited by a user in Kung.

4. Claims 48 and 57 are rejected under 35 U.S.C. 103(a) as being unpatentable over Kung in view of Naudus (US006259691B1) and Kato (US005844918A).

- In regards to Claim 48 and 57,

Kung discloses a method for controlling a voice-over-IP (VOIP) call (Figs. 1-4; Col. 7, lines 15-25; Col. 8, lines 9-13; claim 48 – method for controlling a VOIP call).

Referring to Fig. 1, Kung shows that a call from PSTN 160 may interface IP network 120 through a gateway, where it would be converted to a packetized call (claim 48 – call over circuit-switched network; claim 48 – packetizing the call at a gateway connected to a packet network).

Kung discloses the ability to change quality of service, required bit rate, priority, cost, etc. in real time in response to user input (Col. 7, lines 27-30; claim 57 - optimizing call according to call cost selection). Telephone units are shown to be connected to the system for use as an input device by the user, including DTMF sensing logic (Fig. 3; Col. 23, lines 45-51).

Kung does not explicitly disclose DTMF input indicating a delay associate with the call.

Naudus discloses a method for efficiently transporting DTMF tones in a network-based telephone system. Naudus shows that DTMF input by a user imposes delay in a call (Abstract; claim 48 – DTMF input indicating delay associated with the call; claim 48 – changing packet payload size in response to DTMF).

It would have been obvious to one of ordinary skill in the art at the time of the invention to modify the method of Kung by indicating a delay associated with a call by detecting DTMF input, as shown by Naudus. DTMF input requires further encoding of

the call which adds delay, which may then need to be compensated for within the network.

Kung does not explicitly disclose dynamically adjusting packet payload length to compensate for delay.

Kato discloses a digital transmission/receiving method and apparatus. Kato discloses that adjustments packet length impact the quality and delay of transmission/reception in a system (Col. 1, lines 27-42; claim 48 - adjusting FEC and packet payload length to compensate for delay).

It would have been obvious to one of ordinary skill in the art at the time of the invention to enable adjustments to the FEC and packet length of a transmission as part of the dynamically varying adaptation schemes of Kung, as shown by Kato, because the FEC and packet length of a transmission effect the quality of a transmission. Therefore, adjustments to the FEC and packet length could result in quality changes requested by a user in Kung.

5. Claims 49, 51-53, and 55 are rejected under 35 U.S.C. 103(a) as being unpatentable over Kung in view of Kato.

- In regards to Claim 49, 51-53, and 55,

Kung discloses a method for controlling a voice-over-IP (VOIP) call (Figs. 1-4; Col. 7, lines 15-25; Col. 8, lines 9-13; claim 49 – method for controlling a VOIP call).

Kung shows that VOIP call packets traveling through the IP network may be given a priority to maintain certain QoS requirements (Col. 7, lines 21-25; claim 49 – tracking adaptation schemes used for transmitting packet in a VOIP call).

Kung discloses the ability to change quality of service, required bit rate, priority, cost, etc. in real time in response to user input (Col. 7, lines 27-30; claim 49,53 – monitoring a user response that requests a different level, increase or decrease, of user perceived sound quality for the VOIP call; claim 55 – optimizing call according to call cost selection).

Kung further discloses that the real time changes to the VOIP call may be flexibly performed with regard to congestion in the network (Col. 7, lines 30-35; Col. 17, lines 55-59; claim 51 – dynamically varied according to network congestion).

Kung does not explicitly disclose dynamically adjusting FEC and packet payload length for changing the quality of service.

Kato discloses a digital transmission/receiving method and apparatus. Kato discloses that adjustments to FEC and packet length impact the quality of transmission/reception in a system (Col. 1, lines 27-42; claim 49,51,52,55 - dynamically adjusting FEC and packet payload length for changing quality of service in network; claim 53 – increasing packet length in response to user input).

It would have been obvious to one of ordinary skill in the art at the time of the invention to enable adjustments to the FEC and packet length of a transmission as part of the dynamically varying adaptation schemes of Kung, as shown by Kato, because the

FEC and packet length of a transmission effect the quality of a transmission. Therefore, adjustments to the FEC and packet length could result in quality changes made through requests by a user, network congestion, or cost in Kung.

Allowable Subject Matter

6. Claims 13 and 15-47 are allowed.

- Regarding claim 13,

The prior art of record does not teach or fairly suggest controlling the codec used to encode a call while the call is in progress based upon a user-defined codec selection and determined packet loss ratio compared to a predetermined packet loss ratio.

- Regarding claim 24,

The prior art of record does not teach or fairly suggest controlling the codec used to encode a call while the call is in progress based upon a user-defined codec selection and determined jitter compared to a predetermined jitter.

- Regarding claim 36,

The prior art of record does not teach or fairly suggest controlling the codec used to encode a call while the call is in progress based upon a user-defined codec selection and comparison of a first and second packet loss rate.

- Regarding claim 37,

The prior art of record does not teach or fairly suggest controlling the codec used to encode a call while the call is in progress based upon a user-defined codec selection and measured network congestion compared to predetermined threshold.

7. Claim 54 is objected to as being dependent upon a rejected base claim, but would be allowable if rewritten in independent form including all of the limitations of the base claim and any intervening claims.

The prior art of record does not show a first packet payload type having 20 bytes when the delay is less than a predetermined threshold and a second packet payload type having at least 40 bytes when the delay is greater than the predetermined threshold.

Response to Arguments

8. Applicant's arguments filed 5/23/2006 have been fully considered but they are not persuasive.

- In the Remarks on pg. 15 of the Amendment, Applicant contends that Kung does not disclose changing a codec in the middle of a call in response to signaling from a caller over a network.
- The rejection of claims 1-12, 48-53 and 55-57 rely on the disclosure of Kung to show that various call settings may be changed in the middle of a call in response to signaling form a caller over a network. To meet the explicit limitation of changing codec, the disclosure of Havens is relied upon. The combination of Kung and Havens, therefore, properly meets the limitations contested by Applicant.

- In the Remarks on pg. 18 of the Amendment, Applicant contends that neither Kung nor Kato teaches dynamically varying packet payload length to correspond with a requested level of user perceived sound quality.
- As shown above, the rejection of claim 49 relies on the disclosure of Kung to show that various call settings may be changed in the middle of a call in response to signaling form a caller over a network. To meet the explicit limitation of varying packet payload length, the disclosure of Kato is relied upon. The combination of Kung and Kato, as shown above, properly meets the limitations contested by Applicant.

Conclusion

9. The prior art made of record and not relied upon is considered pertinent to applicant's disclosure.

- Oran (US006775265B1)
- Ogier (US 20030095504A1)

10. Applicant's amendment necessitated the new ground(s) of rejection presented in this Office action. Accordingly, **THIS ACTION IS MADE FINAL**. See MPEP § 706.07(a). Applicant is reminded of the extension of time policy as set forth in 37 CFR 1.136(a).

A shortened statutory period for reply to this final action is set to expire THREE MONTHS from the mailing date of this action. In the event a first reply is filed within TWO MONTHS of the mailing date of this final action and the advisory action is not mailed until after the end of the THREE-MONTH shortened statutory period, then the shortened statutory period will expire on the date the advisory action is mailed, and any extension fee pursuant to 37 CFR 1.136(a) will be calculated from the mailing date of the advisory action. In no event, however, will the statutory period for reply expire later than SIX MONTHS from the date of this final action.

Any inquiry concerning this communication or earlier communications from the examiner should be directed to Gregory B. Sefcheck whose telephone number is 571-272-3098. The examiner can normally be reached on Monday-Friday, 8:00am-4:30pm.

If attempts to reach the examiner by telephone are unsuccessful, the examiner's supervisor, Seema Rao can be reached on 571-272-3174. The fax phone number for the organization where this application or proceeding is assigned is 571-273-8300.

Information regarding the status of an application may be obtained from the Patent Application Information Retrieval (PAIR) system. Status information for published applications may be obtained from either Private PAIR or Public PAIR. Status information for unpublished applications is available through Private PAIR only. For more information about the PAIR system, see http://pair-direct.uspto.gov. Should you have questions on access to the Private PAIR system, contact the Electronic Business Center (EBC) at 866-217-9197 (toll-free). If you would like assistance from a USPTO Customer Service Representative or access to the automated information system, call 800-786-9199 (IN USA OR CANADA) or 571-272-1000.

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8-18-2006

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